IN THE SPECIFICATION

Please amend the paragraph beginning on page 1, line 6 as follows:

This invention relates to an information management method and an information management apparatus for ensuring the compatibility of [[a]] recording medium storing signals that are coded by different methods.

Please amend the paragraph beginning on page 3, line 2 as follows:

A1

"Polyphase Quadrature Filters - A New Subband Coding Technique", Joseph H. Rothweiler, ICASSP 83, ~~BOSTON) described~~ BOSTON describes a band splitting technique using a PQF. The PQF described in the above paper is devised to utilize the phenomenon that, if the signals that are subjected to subband coding using the PQF are thinned out to show a signal rate corresponding to the related bandwidth and consequently aliasing noises are generated between adjacent subbands, the generated aliasing noises are cancelled by the aliasing noises that are generated between adjacent subbands in the subsequent band synthesis. Therefore, again, the coding loss can be substantially eliminated by using a PQF as time splitting filter so long as the signals of each subband are coded with a satisfactory level of accuracy.

Please amend the paragraph beginning on page 3, line 13 as follows:

A2

Spectrum transform techniques include those adapted to split the input audio signals into blocks on the basis of a predetermined time unit (frame) and transform the signals on a time base into those on a frequency base by subjecting them to discrete Fourier transform (DFT), discrete cosine transform (DCT) or modified discrete cosine transform (MDCT) on a block by block bases. For MDCT, refer to "Subband/Transform Coding Using Filter Bank Designs Based on

A2 Time Domain Aliasing Cancellation", J.P. Princen, A. B. Bradley, Univ. of Surrey Royal Melbourne Inst. Of Tech. ICASSP ~~1987)~~ 1987.

Please amend the paragraph beginning on page 4, line 19 as follows:

A3 Generally, the frequency resolution is enhanced to give rise to a phenomenon of concentration of energy on a specific spectrum signal component if the transform blocks for spectrum transform are made long. Therefore, a coding operation can be conducted more efficiently by using MDCT than by using DFT or DCT because, if a long transform block length is used for spectrum transform with MDCT, a half of the total number of sample data are made to overlap between two adjacent transform blocks and the number of the obtained spectrum signal components is not increased relative to the number of the original sample data on the time base. Additionally, the connection distortion between transform blocks of waveform signals can be alleviated by causing adjacent transform blocks to overlap by a sufficiently long span. However, it should be noted that a long transform block means that more work areas are required for the transform to possibly baffle the efforts for down-sizing the signal reproduction means. Particularly, the use of a long transform ~~blocks~~ block can entail a cost rise when it is difficult to raise the degree of integration of semiconductors.

Please amend the paragraph beginning on page 5, line 14 as follows:

A4 Meanwhile, with the above described technique of splitting the signal frequency bands by means of a filter and spectrum transform, the quantization noise generation band can be limited when quantizing the signal components obtained by the band division. In other words, it is possible to perform a coding operation highly efficiently in terms of the auditory perception by limiting the quantization noise generation band, typically utilizing the masking effect. The

A4 masking effect refers to an effect that a large sound hides a small sound to the ears. Thus, the signal sound itself can be made to hide the quantization noise generated as a result of quantization due to the masking effect. Therefore, if audio signals are compressed in a way that maximally exploits the masking effect, the sound reproduced from the audio signals obtained by expanding the compressed audio signals will be almost the same as the original sound to the ears in terms of sound quality. However, it should be noted that the generation of quantization noise has to be controlled in terms of both time and frequency in order to maximally exploit the masking effect. More specifically, the masking effect can vary along the time base in terms of the duration of the effect and as far as an attack where the signal level abruptly rises from a relatively low level to a high level is concerned, the masking effect works only several milliseconds temporally before the attack whereas it works for a considerably long time after the attack. Therefore, assuming a transform block containing an attack and low level signals located before and after the attack, if a low level signal is found for more than several milliseconds temporally before the attack and the level of the quantization noise generated in the transform block is higher than that of the low level signal, the level of the quantization noise generated in the transform block exceeds that of the low level signal (and hence is not hidden by the small sound of the low level signal) so that there arises a phenomenon of so-called pre-echo that is very harsh to the ears.

Please amend the paragraph beginning on page 7, line 17 as follows:

A5 Additionally, the operation of coding the data of the subbands obtained by frequency splitting is preferably carried out by allocating a predetermined number of bits or by adaptively allocating an appropriate number of bits to each of the subbands (bit allocation). For instance,

A5 the technique of adaptively allocating an appropriate number of bits to the MDCT coefficient data of each subband obtained by MDCT conducted on each transform block will be used for the operation of coding the coefficient data obtained by MDCT.

Please amend the paragraph beginning on page 8, line 4 as follows:

A6 "Adaptive Transform Coding of Speech Signals", R. Zelinski and P. Noll, IEEE Transactions of Acoustics, Speech and Signal Processing, Vol. ASSP-25, No. 4, August ~~1997)~~ ~~described~~ 1997 describes a technique of bit allocation based on the signal size of each subband. However, while a flat quantization noise spectrum is produced to minimize the noise energy with this technique, the actual feeling of hearing noise is not optimal to the auditory sense because it does not utilize the masking effect.

Please amend the paragraph beginning on page 10, line 12 as follows:

A7 There is also known a technique of determining the quantization accuracy information typically from the normalization coefficient information in a decoder instead of directly encoding the quantization accuracy information. However, with this technique, the relationship between the normalization coefficient information and the quantization accuracy information becomes fixed when the standards are installed so that it is no longer possible to introduce an improved system for controlling the quantization accuracy on the basis of an enhanced auditory model in the future.

Please amend the paragraph beginning on page 11, line 11 as follows:

A8 Note that any of the above listed coding techniques is applicable to each channel of an acoustic signal constituted by a plurality of channels. For instance, any of them may be applied separately to the L channel that corresponds to the left-side loudspeaker and also to the R

channel that corresponds to the right-side loudspeaker. Furthermore, any of them may be applied to the (L+R) / 2 signal obtained by adding the signal of the L channel and that of the R channel or both of the (L+R) / 2 and (L-R) / 2 signals for efficient coding. For example, Japanese Patent Application Laid-Open No. 10-336039 filed by the applicant of this patent application proposes in its specification and drawings a method of reducing the bandwidth of the (L-R) / 2 signal relative to the that of the (L+R) / 2 signal, paying attention to the fact that the feeling of stereophony is dominantly affected by low frequency side signals. With this technique, it is possible to efficiently ~~carrying~~ carry out a coding operation, using a reduced number of bits, while maintaining the feeling of stereophony as perceived by the auditory sense. It should be noted here that, since the amount of data required for coding signals of a channel is half ~~is a half~~ of that of data for coding signals of two channels independently, a technique of establishing a set of standards providing both a mode for recording monaural signals of a single channel and a mode for recording stereo signals of two channels is popularly used so that signals may be recorded as monaural signals when a long recording time is expected for recording signals on a recording medium.

A9

Please amend the paragraph beginning on page 12, line 12 as follows:

A10

As described above, novel techniques for improving the coding efficiency have been developed almost incessantly so that, if a set of standards accommodating a newly developed coding technique is used, it will normally be possible to record signals for a prolonged period of time on an information recording medium or, if the recording time is the same, record higher quality audio signals.

Please amend the paragraph beginning on page 12, line 18 as follows:

A11

When establishing a new set of standards, provisions are normally made to accommodate possible revisions and/or extensions in the future so that flag information and other necessary pieces of information relating to the standards may be recorded on the recording medium in advance. For instance, a 1-bit flag information of "0" may be recorded on the recording medium when the standards are established for the first time and the flag information may be turned to "1" when the standards are revised. With this arrangement, an apparatus that is adapted to the revised standards checks if the flag information recorded on the recording medium is equal to "0" or "1" and reads and reproduces signals from the information recoding medium according to the revised standards if the flag information is "1", whereas it reads and reproduces signals from the information recording medium according to the original standards if the flag information is "0" and the apparatus is not adapted to the original standards.

Please amend the paragraph beginning on page 13, line 13 as follows:

A12

However, if apparatus that can reproduce signals ~~signal~~ that are recorded according to a set of standards (which is to be referred to as "the old standards" or "the first coding system" hereinafter) become popular and widely used and a new set of standards accommodating a more efficient coding system[["]], which may be superceding standards, (which is to be referred to as "the new standards" or "the second coding system" hereinafter) is established, the users of the apparatus will have to experience the inconvenience of not being able to replay any information recording medium where signals are recorded according to the new standards. Apparatus that can reproduce and/or record signals according to the old standards will be referred to as apparatus adapted to the old standards hereinafter.

Please amend the paragraph beginning on page 14, line 3 as follows:

A13

Particularly, there may be apparatus that are adapted to the old standards and try to reproduce all the signals recorded on the information recording medium as if they are coded according to the old standards, disregarding the flag information recorded on the information recording medium. In other words, if the information recording medium stores signals coded according to the new standards, the apparatus adapted to the old ~~olde~~ standards cannot recognize it. Then, if the apparatus adapted to the old standards tries to reproduce signals recorded according to the new standards as if they are signals recorded according to the old standards, the apparatus may not operate properly and/or give rise to terrible noises.

Please amend the paragraph beginning on page 14, line 18 as follows:

A14

On the other hand, Japanese Patent Application Laid-Open No. 10-302405 filed by the applicant of the present patent application proposes a technique with which an apparatus adapted to the old standards can reproduce signals coded according to the old standards if the recording medium stores both signals ~~signal~~ coded according to the old standards and those coded according to the new standards while an apparatus adapted to the new standards can reproduce from the recording medium both signals coded according to the old standards and those coded according to the new standards and any possible degradation of signal quality that can arise when signals coded according to different sets of standards are recorded on a same information recording medium can be minimized. Note that, in the following description, an apparatus that can reproduce and/or record signals coded according to the new standards, which may be superceding standards, is referred to as apparatus adapted to the new standards.

Please amend the paragraph beginning on page 15, line 17 as follows:

A15 To be more accurate, while management data (so-called TOC) including track replay mode information, start address information and end address information have to be stored in the management data area of the recording medium defined by the old standards so that they may be referred to by an apparatus adapted to the old standards, data on the additional information (extended information) such as the information on the replay mode adapted to the new standards and necessary for an apparatus adapted to the new standards to reproduce value-added data have to be stored in an area (extended management data area) that can be referred to only by an apparatus adapted to the new standards so that they may ~~by~~ not be referred to nor erased by an apparatus adapted to the old standards.

Please amend the paragraph beginning on page 24, line 6 as follows:

A16 The optical head 53 typically comprises a laser beam source such as a laser diode, a collimator lens, an objective lens, a polarization beam splitter, a cylindrical lens and other optical components along with a photodetector having a light receiving section showing a predetermined pattern. The optical head 53 is arranged at a position opposite to said magnetic head 54 with the magneto-optic disk 1 interposed therebetween. When recording data on the magneto-optic disk 1, a modulated magnetic field is applied to the data to be recorded by driving the magnetic head 54 by means of head drive circuit 66 of the recording system of the apparatus which will be described hereinafter, while the target track of the magneto-optic disk 1 is irradiated with a laser beam emitted from the optical head 53 for thermo-magnetic recording in a magnetic modulation mode. The optical head 53 is adapted to detect the reflected beam of the laser beam irradiating the target track and also detect any ~~focussing~~ focusing errors by means of the so-called

astigmatism technique and any tracking errors by means of the so-called push-pull technique. When reproducing data from the magneto-optic disk 1, the optical head 53 detects ~~focussing~~ focusing errors and tracking errors and, at the same time, detects the difference in the polarization angle of the laser beam (color rotational angle) reflected from the target track.

Please amend the paragraph beginning on page 25, line 9 as follows:

A17 The servo control circuit 56 typically comprises a ~~focussing~~ focusing servo control circuit, a tracking servo control circuit, a spindle motor servo control circuit and a sled servo control circuit. The ~~focussing~~ focusing servo control circuit controls the ~~focussing~~ focusing operation of the optical system of the optical head 53 so as to reduce the ~~focussing~~ focusing error signal to nil. The tracking servo control circuit controls the tracking operation of the optical system of the optical head 53 so as to reduce the tracking error signal to nil. The spindle motor servo control circuit controls the spindle motor 51 so as to make it drive the magneto-optic disk 1 to rotate at a predetermined rotational speed (e.g., at a constant linear speed). The sled servo control circuit moves the optical head 53 and the magnetic head 54 to the target track of the magneto-optic disk 1 as specified by system controller 57. Thus, the servo control circuit 56 is adapted to perform various control operations and transmit information indicating the operations of various components that are controlled by the servo control circuit 56 to the system controller 57.

Please amend the paragraph beginning on page 26, line 15 as follows:

A18 The reproduction time is obtained by multiply the address information that is reproduced from the recording track of the magneto-optic disk 1 on a sector by sector basis by using the header time on a sector by sector bases and the subcode Q data by the reciprocal number of the

data compression ratio (e.g., 4 if the compression ration is 1/4) and displayed on the display section 59. If the absolute time information is recorded on the recording track of the magneto-optic disk (1) (and hence the disk is pre-formatted), it is also possible to read the absolute time information on the pre-formatted disk and multiply it by the reciprocal number of the data compression ratio so as to display the current position in terms of the actual recording time.

A18

Please amend the paragraph beginning on page 31, line 4 as follows:

A19

The memory 72 is controlled by the system controller 57 for the operation of writing data in and reading data from it. The reproduced data fed the decoder 71 at a transfer rate of 75 sectors/sec. are written in it at the same transfer rate of 75 sectors/sec. in a burst-like fashion. The reproduced data that are written in the memory 72 at a transfer rate of 75 sectors/sec. in a burst-like fashion are then read out continuously from it at a transfer rate of 9.375 sectors/sec. that corresponds to the data compression ratio of 1/8.

Please amend the paragraph beginning on page 33, line 4 as follows:

A20

Referring to FIG. 3, at the converter 111a, the input signal 120a is split into two subbands by a band splitting filter 112a and the subband signals 120b, 120c obtained by splitting the input signal are respectively transformed into spectrum signal components 120d, 120e by respective forward converters 112b, 112c. Note that the input signal 120a corresponds to the signal waveform 110a in FIG. 2 and the spectrum signal components 120d, ~~120e~~ 120e correspond respectively to the signal frequency components 110b in FIG. 2. In the converter 111a having the configuration as illustrated in FIG. 3, the bandwidth of each of the signals 120b, 120c obtained by splitting into two subbands is equal to 1/2 of that of the input signal 120a. In other words, the input signal 120a is decimated to 1/2. It may be needless to say that many alternative

configurations are conceivable for the converter 111a. For example, it may be so arranged that the input signal is directly transformed into a spectrum signal by MDCT. A transform technique such as DFT or DCT may be used instead of MDCT for the purpose of the invention. While it is also possible to split a signal into subband components by means of a band splitting filter, it is advantageous for the purpose of the present invention to use a technique of transforming a signal into frequency components because many frequency components can be obtained with a limited number of arithmetic operations when an information coding method according to the invention is used.

A20

Please amend the paragraph beginning on page 37, line 10 as follows:

A21

The number of bits actually required to restore the spectrum signals of the transform blocks is determined as a function of the number of coding units used fro the coding operation and the number of quantization bits used for the quantization accuracy information of the coding units and may vary from frame to frame. Only the number of bits required to restore the spectrum signals as counted from the top are significant and any remaining area of the frame is idle area and hence does not affect the reproduced signals. Normally, the idle area of each frame is minimized so as to effectively utilize as many ~~as~~ bits ~~and improve the sound quality.~~

Please amend the paragraph beginning on page 43, line 10 as follows:

A22

Referring to the signal component decoder 114b of FIG. 15, the code 140b sent from the code string decomposer 114a includes a tone component data 180a and a non-tone-related signal component 180b and the data and the signal component are respectively sent to tone component decoder 118a and non-tone component decoder 118b. The tone component decoder 118a decodes the tone-related signal component ~~form~~ from the tone component data as shown in FIG.

A22

13 and outputs the obtained tone-related signal component 180c. On the other hand, the non-tone component decoder 118d decodes the non-tone related signal component and outputs the obtained non-tone related signal component 180d. The tone-related signal component 180c and the non-tone-related signal component 180d are then sent to spectrum signal synthesizer 118c. The spectrum signal synthesizer 118c synthetically combines the tone-related signal component and the non-tone-related signal component according to the position data and outputs the obtained signal component 180e. Note that the configuration of the tone component decoder 118a and the non-tone component decoder 118b is same as that of the circuit of FIG. 7 in terms of signal decoding.

Please amend the paragraph beginning on page 48, line 20 as follows:

A23

In view of this problem, the applicant of the present patent application proposes in the specification and the drawings of Japanese Patent Application Laid-Open No. 10-302405 a technique with which, when both signals coded according to A codec (the old standards) and those coded according to B codec (the new standards) are recorded on a same disk, the signals according to A codec can be reproduced by an apparatus adapted to the old standards while an apparatus adapted to the new standards can reproduce any signals recorded on the disk regardless if they are coded according to A codec or B codec and, additionally, the risk of gradation of signal quality that ~~cay~~ can arise by recording signals coded according to different standards can be alleviated. When signals according to old standards (A codec) and those according to the new standards (B codec) are recorded on a same disk, the storage area allocated to each type of signals is inevitably reduced so that consequently it may be difficult to maintain the quality level of signals when they are reproduced. However, the technique disclosed in the specification and

A22

the drawings of Japanese Patent Application Laid-Open No. 10-302405 can also alleviate the degradation of sound quality.

Please amend the paragraph beginning on page 61, line 2 as follows:

A24

In the instance of FIG. 19, the address information is handled as a unit and includes the start addresses and the end addresses as well as the mode information (to be referred to as track mode hereinafter) and link information (to be referred to as link pointer hereinafter) for the signals in each of the areas indicated by a start address and an end address (the unit or its recording areas may be called as a slot(s)). As track mode, the number of audio channels (e.g., the number of channel(s) for monaural signals or stereo signals) of the recording areas, ~~the rewrite protection flag and~~ the flag indicating if digital signals are recorded or not, and the rewrite protection flag ~~not~~ are ~~recorded~~ stored. The link pointer is typically used when a tune is recorded in two areas that are physically remote from each other so that it links to the areas that are physically remote from each other so that it links the areas to restore the tune and stores information on the address storing positions. If there is no need of link, 0 will be recorded.

Please amend the paragraph beginning on page 61, line 14 as follows:

A25

Information on the idle address storing positions is stored at address 11 of the management data area as information showing the top of each idle slot. Idle slots are connected to each other by means of the link arranged in each slot and the link of the last slot is made equal to 0. ~~TInformation~~ Information on idle area address storing positions is stored at address 12 of the management data area. In other words, they represent the slots where the addresses of the idle areas (unused areas) on the disk. When a plurality of idle areas exist on the disk, they are connected by means of the link arranged in each slot. If the number of recordable areas on the

A25 disk is found to be less than the number of areas that the recording apparatus can use or equal to 0, the information on idle area address storing positions is made equal to 0.

Please amend the paragraph beginning on page 67, line 6 as follows:

A26 While the positions storing different pieces of information in the extended management data area are same as those of their counterparts in the management data area as viewed from the top thereof, other embodiments of the present invention where the former positions are different from the corresponding positions in the management data area may feasibly <u>be</u> realized.

Please amend the paragraph beginning on page 67, line 18 as follows:

A27 While FIG. 21 shows an operation of recording strings of codes adapted to the new standards, if ~~there is~~ no code string adapted to the new standards exists on the recording medium when an apparatus adapted to the new standards is used to record strings of codes adapted to the old standards, it is possible to record them by means of the apparatus adapted to the new standards, using only the existing management data area. On the other hand, it is also possible to an apparatus adapted to the new standards to handle both management data and extended management data by using both the management data <u>area</u> ~~rea~~ and the extended management data are regardless if the strings of code to be recorded are adapted to the old standards or the new standards.

Please amend the paragraph beginning on page 69, line 3 as follows:

A28 If, on the other hand, it is determined in Step S202 that the management data extension <u>flag</u> ~~flat~~ is not equal to 0, the compressed data recording/reproduction apparatus determines in Step S203 if the track mode of the position corresponding to the specified track mode in the extended management data that is stored in the extended data area is equal to 0 or not.

Please amend the paragraph beginning on page 69, line 18 as follows:

A29

While the strings of codes adapted to the new standards are reproduced ~~by an apparatus adapted to the new standards in the track mode according to the new standards~~ in the track mode according to the new standards in the illustration of FIG. 22, it may alternatively be arranged so ~~arrangement~~ that the apparatus adapted to the new standards can select either the track mode according to the new standards or the track mode according to the old standards.

Please amend the paragraph beginning on page 73, line 13 as follows:

A30

While the present invention is described above in terms of audio signals, the method according to the invention can be applied to occasions where the signals reproduced by an apparatus adapted to old standards are video signals. Additionally, while the present invention is described above in terms of coded bit streams on a recording medium, the method according to the invention can be applied to transmission of bit streams. Finally, the recording medium is not ~~no~~ limited to those listed above and a semiconductor memory can also be used as recording medium for the purpose of the present invention.